

From the  
INTERNATIONAL PRELIMINARY EXAMINING AUTHORITY

To:

CHRISTENSEN, Mikael T.  
OTICON A/S  
Strandvejen 58  
DK-2900 Hellerup  
DANEMARK

PCT

NOTIFICATION OF TRANSMITTAL OF  
THE INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT

(PCT Rule 71.1)

Date of mailing  
(day/month/year) 07.11.2001

Applicant's or agent's file reference  
P-1999-014-1

IMPORTANT NOTIFICATION

International application No.  
PCT/DK00/00379

International filing date (day/month/year)  
07/07/2000

Priority date (day/month/year)  
19/07/1999

Applicant  
OTICON A/S et al.

1. The applicant is hereby notified that this International Preliminary Examining Authority transmits herewith the international preliminary examination report and its annexes, if any, established on the international application.
2. A copy of the report and its annexes, if any, is being transmitted to the International Bureau for communication to all the elected Offices.
3. Where required by any of the elected Offices, the International Bureau will prepare an English translation of the report (but not of any annexes) and will transmit such translation to those Offices.

4. REMINDER

The applicant must enter the national phase before each elected Office by performing certain acts (filing translations and paying national fees) within 30 months from the priority date (or later in some Offices) (Article 39(1)) (see also the reminder sent by the International Bureau with Form PCT/IB/301).

Where a translation of the international application must be furnished to an elected Office, that translation must contain a translation of any annexes to the international preliminary examination report. It is the applicant's responsibility to prepare and furnish such translation directly to each elected Office concerned.

For further details on the applicable time limits and requirements of the elected Offices, see Volume II of the PCT Applicant's Guide.

Name and mailing address of the IPEA/

 European Patent Office  
D-80298 Munich  
Tel. +49 89 2399 - 0 Tx: 523656 epmu d  
Fax: +49 89 2399 - 4465

Authorized officer

Cornudet-Henschel, V

Tel. +49 89 2399-7371



(19) World Intellectual Property Organization  
International Bureau



(43) International Publication Date  
25 January 2001 (25.01.2001)

PCT

(10) International Publication Number  
**WO 01/06746 A2**

(51) International Patent Classification<sup>7</sup>: H04M 9/00

(DK). EKELID, Michael [DK/DK]; Oticon A/S, Strandvejen 58, DK-2900 Hellerup (DK).

(21) International Application Number: PCT/DK00/00379

(22) International Filing Date: 7 July 2000 (07.07.2000)

(74) Common Representative: OTICON A/S; Mikael, T., Christensen, Strandvejen 58, DK-2900 Hellerup (DK).

(25) Filing Language: English

(81) Designated States (*national*): AU, BR, CA, JP, US.

(26) Publication Language: English

(84) Designated States (*regional*): European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

(30) Priority Data:  
PA 1999 01043 19 July 1999 (19.07.1999) DK

(71) Applicant (*for all designated States except US*): OTICON A/S [DK/DK]; Strandvejen 58, DK-2900 Hellerup (DK).

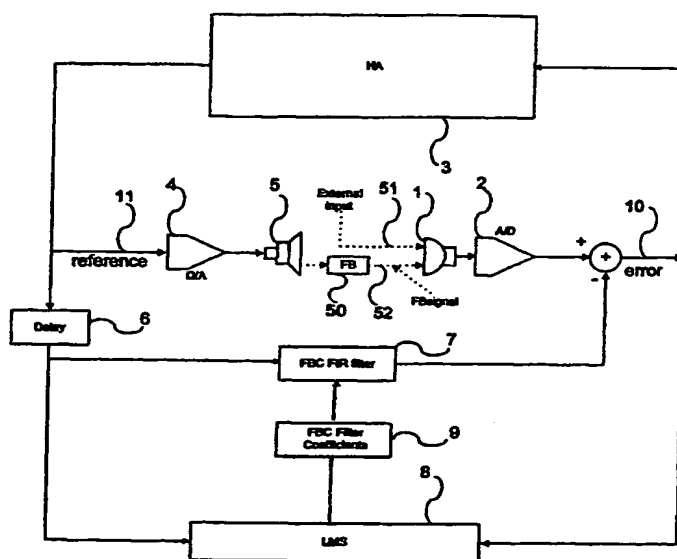
**Published:**  
— Without international search report and to be republished upon receipt of that report.

(72) Inventors; and

(75) Inventors/Applicants (*for US only*): NIELSEN, Jakob [DK/DK]; Oticon A/S, Strandvejen 58, DK-2900 Hellerup

*For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.*

(54) Title: FEEDBACK CANCELLATION USING BANDWIDTH DETECTION



(57) Abstract: The invention relates to a method for cancelling feedback in an acoustic system comprising a microphone, a signal path, a speaker and means for detecting presence of feedback between the speaker and the microphone, the method comprising providing an LMS algorithm for processing the signal; where the LMS algorithm operates with a predetermined adaptation speed when feedback is not present; where the LMS algorithm operates an adaptation speed faster than the predetermined adaptation speed when feedback is present, and where the means for detecting the presence of feedback is used to control the adaptation speed selection of the LMS algorithm, where the feedback detection means comprises bandwidth detection means for determining the presence of a feedback signal.

WO 01/06746 A2

## TITLE

## Feedback cancellation using bandwidth detection

5

## TECHNICAL FIELD

The present invention concerns hearing aids. In many hearing aids, for example "In-the-ear" (ITE) and "Behind-the-ear" (BTE), the microphone and the receiver (telephone) components are placed close to each other. This may result in that  
10 the sound produced by the receiver leaks back into the microphone. This may occur when the hearing aid shell or the ear mould does not fit sufficiently tight in the ear canal. Given enough amplification in the hearing aid, the loop gain of the system may exceed 0 dB at some frequency and a feedback oscillation may be produced.

15

## BACKGROUND OF THE INVENTION

The present invention is based on algorithms previously proposed in the literature. The invention concerns a number of algorithm modifications, which  
20 overcome some of the limitations of other systems used for feedback reduction in hearing aids.

The invention relates to a feedback cancellation algorithm, which does not need an artificial noise signal in order to estimate the feedback transfer function. The  
25 input signal received from the environment, or the feedback oscillation signal, is used to drive the estimation process. In this fashion, the hearing aid user does not listen to an added noise signal, and a higher sound quality is possible. However, it is well known that such 'no-noise' algorithms can have audible side effects under certain circumstances, especially when environmental signals with long  
30 autocorrelation functions are present at the microphone.

The autocorrelation function for a signal describes the average correlation between two signal values, which are separated by a time difference "Lag". In loose terms, the autocorrelation function describes how "predictable" a signal

value is, given the other samples in the signal. Some signals, for example periodic signals, are highly predictable and, correspondingly, the autocorrelation function does not vanish even for large values of Lag. Other signals, such as white noise, are very little predictable, and their autocorrelation function quickly vanish for  
5 increasing values of Lag. For signals with a long autocorrelation function, a future sample value can be predicted with a high degree of confidence, given the past samples. In other words, new samples of the signal do not provide much new information. Careful analysis of feedback cancellation systems reveal that signals with long autocorrelation may drive the adaptive system to produce poor  
10 estimates of the feedback path.

It is the objective of the present invention to provide a method and a hearing aid for feedback cancellation, which improves the result of the feedback cancellation by having fewer audible side effects and thereby gives an improved user comfort.  
15

#### SUMMARY OF THE INVENTION

According to the invention the objective is achieved by a method, which comprises the feature of claim 1. According to the invention the first objective is  
20 likewise achieved by a hearing aid, which comprises the features of claim 6.

Hereby a low adaptation speed, which generally improves the sound quality for signals with long autocorrelation functions, is applied when no feedback oscillation is present and a high adaptation speed, which is desirable to reduce  
25 feedback oscillations quickly, is applied when feedback oscillation is present hereby maintaining the preferred mode when feedback is not present and quickly changing the mode essentially without audible oscillations. This results in fewer audible side effects and an improved user comfort.

30 Advantageous embodiments are described in the dependent claims 2-5 and 7. The function and effect of these is explained in connection with the preferred embodiment.

## BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a schematic diagram showing a previously known system used for feedback cancellation;

5

FIG. 2 is a schematic diagram showing an embodiment of the system according to the present invention.

FIG. 3 is a schematic diagram showing the feedback detection system according to the invention;

10

## DETAILED DESCRIPTION OF THE DRAWINGS

A well-known principle for feedback cancellation in hearing aids is shown in fig. 1. All the components described below, except blocks (1), (5) and (50), operate in the discrete time domain.

15

The components are as follows: (1) is a microphone which picks up the sound from the environment (51) ("External input") and the feedback signal (52) ("FBSignal"); (2) is a microphone amplifier and an analog-to-digital converter (A/D); (3) is the hearing aid amplifier with filters, compressors, etc.; (4) is a digital-to-analog converter and a power amplifier; (5) is the hearing aid receiver; (50) is the acoustic feedback path (outside the hearing aid); (6) is a delay unit whose delay matches the delay through the components (4), (5), (50), (1) and (2). (7) is an N-tap finite impulse response (FIR) filter which is intended to simulate the combined impulse response of components (4), (5), (1), (2) and (50). (8) is an adaptive algorithm which will adjust the coefficients (9) of the filter (7) so as to minimize the power of the error signal (10).

20

25

The algorithm (8) is well known as the Least Mean Square (LMS) algorithm. The algorithm requires a reference signal (11), which is used to excite the path consisting of the components (4), (5), (1), (2) and (50). The correlation between the reference signal (11) and the error signal (10) is used to compute the adjustment of the coefficients (9).

30

The system utilizes the output signal (11) from the hearing aid amplifier block (3) as a driving signal for the LMS algorithm, thereby eliminating the need for a disturbing noise in the receiver (5).

5

For some external input signals, the LMS based algorithm used in the application shown in fig. 1 is known to have difficulty adjusting the coefficients (9) as desired, i.e. to match the path consisting of components (4), (5), (1), (2) and (50). The difficulties are greatest for signals with long autocorrelation functions.

10 Mismatched coefficients may lead to audible side effects, which can be very disturbing to a hearing aid user. Those may comprise audible oscillations and change in gain characteristics and frequency characteristics. One general remedy against this problem is to use a low adaptation speed, but this leads to poorer performance of the system because the coefficients cannot track changes in the  
15 acoustic feedback path (50) quickly, resulting in a long feedback cancellation time.

The basic system shown in fig. 1 may be improved in various ways to minimize the side effects associated with certain input signals. Many authors have  
20 proposed additional system blocks, which will improve the sound quality while maintaining an acceptable adaptation speed.

The present invention is based on the system diagram shown in fig. 1, and the invention consists of additional features, which will improve the sound quality  
25 and maintain an acceptable adaptation speed.

FIG.2 shows the block diagram of the general system and the components of the invention.

30 The embodiment shown includes three features: Adaptation rate control, a frequency-selective adaptation procedure, and a feedback oscillation detector.

#### **Adaptation rate control**

Two well known operation modes for the LMS algorithm are the "standard" mode and the "normalized" mode. In the "standard" mode, the coefficients are updated by an amount that depends on the short-term power of the error signal and the reference signal. This means that the update rate is faster when more powerful  
5 signals are processed by the hearing aid. In the "normalized" mode, the update rate is made nearly independent of the signal power, due to a normalization of the update equation.

As described earlier, a low adaptation speed generally improves the sound  
10 quality for signals with long autocorrelation functions. In contrast, a high adaptation speed is desirable to reduce feedback oscillations quickly.

Other authors have previously proposed changing the adaptation rate factor (often known as " $\mu$ ") when feedback oscillations are detected. Although this does  
15 increase the adaptation speed, it also allows coefficients to deteriorate proportionally faster, in those situations where signals with long autocorrelation functions are present at the hearing aid input.

In the present invention, the fact that feedback oscillations often have a high  
20 power is used. In many hearing aids, the output level is limited by compressor circuits, and in many cases the maximum output level is well above the normally used output level, for example when speech and other environmental signal are present. It is therefore assume that the feedback oscillations have a higher power than the environmental signal, in most cases where feedback problems exist.

25

Additionally, the feedback oscillation has the desirable property that its frequency is generally equal to the frequency where the loop gain currently is highest, i.e. where the fastest adaptation is needed.

30 For the reasons mentioned above, it is very effective to utilize the feedback oscillation signal itself as a driving signal for the adaptation.

When the "normalized" adaptation approach is used, the high-power feature of the feedback oscillation is not utilized. If, instead, the "standard" update

approach were used, the high power feature of the feedback oscillation would be utilized. At the same time, however, stronger signals in general would cause a higher adaptation speed, which could lead to more autocorrelation problems.

5 The present invention introduces a new normalization scheme, which will generally maintain the low adaptation speed and the normalized operation mode, except when a feedback oscillation is detected. When a feedback oscillation is detected, the system is switched from normalized operation to standard operation by the switch (13), and the full power of the feedback oscillation signal is  
10 therefore allowed to adapt the coefficients. During "standard" operation, the update parameter (14) is chosen to such a value (53) that the external input (51) produces approximately the same update rate as it would in "normalized" operation. Assuming that the external input signal (51) maintains nearly constant properties before and during the feedback oscillation, the switch of normalization  
15 procedure will be nearly transparent to the external signal (51). This ensures that the sound quality remains high, even though the adaptation speed has been increased due to the higher power in the feedback oscillation. The update parameter (53) to be used during standard mode is estimated in component (12) before the feedback oscillation is detected. During intervals of feedback  
20 oscillations, controls signal (15) prevents (12) from updating the parameter (53).

The switch from normalized mode to standard mode may be controlled by a feedback oscillation detector (49) through its output signal (15). The switch (13) may also be controlled by other conditions, which could result in feedback  
25 oscillations, for example when the acoustic feedback is rapidly decreased.

The adaptive LMS algorithm (8) may be implemented as the following set of equations:

30 Normalized operation:

$$h_k(n+1) = h_k(n) + \frac{a \cdot r(n-k) \cdot e(n)}{b + \sum_p r(n)^2}, \quad p = 1..N$$

$$k=1..N \quad (E1)$$



Standard operation:

$$h_k(n+1) = h_k(n) + \frac{a \cdot r(n-k) \cdot e(n)}{b + LT_{sum}}, \quad k = 1..N \quad (E2)$$

- 5 In these equations,  $h_k(n)$  is the k'th coefficient in the FIR filter at sample time n; a is a constant which determines the general adaptation speed of the algorithm (this constant is sometimes referred to as "μ"); b is a small constant which prevents division by 0 for very small values of the reference signal; N is the number of coefficients in the filter (7); r(n) is the reference signal (30) sample value at time n; e(n) is the error signal (28) sample value at time n; and  $LT_{sum}$  is a value computed as described below.

The sum term of the denominator of E1 is equal to the signal (54).  $LT_{sum}$  is equal to the signal (53).

15

$LT_{sum}$  (equal to (53)), which is computed by component (12), may be updated according to eq. (E3a):

$$LT_{sum}(n+1) = LT_{sum}(n) \cdot \beta_{LT} + SumSq(n) \cdot \alpha_{LT} \quad (E3a)$$

- 20 In equation (E3a) SumSq(n) is defined as follows (E3b):

$$SumSq(n) = \sum_{p=1..N} r(n)^2, \quad p = 1..N$$

(E3b)

$\alpha_{LT}$  and  $\beta_{LT}$  are time constants, which control the length of the exponential window over which the value of  $LT_{sum}$  is computed.

25

Eq. (E3a) should not be updated while a feedback oscillation is present, since  $LT_{sum}$  should reflect the long-term value of SumSq for segments without oscillation. Once the feedback oscillation has disappeared, eq. (E3a) may be updated again.

In E1 and E3b, the reference signal  $r(n)$  is used for normalizing the update equation. However, other signals in the system shown in fig. 2 may also be used instead of  $r(n)$ . In the literature, the error signal  $e(n)$  has been used instead of  $r(n)$  for normalization; and even combinations of  $r(n)$  and  $e(n)$  have been used. The present invention will work for any type of normalization, in which the denominator in E1 and E2 is increased when the power level in the feedback loop consisting of (1), (2), (3), (4), (5) and (50) is increased.

#### 10    **Frequency-selective adaptation**

Many feedback cancellation systems proposed earlier contain some form of frequency weighting of the signals which enter the LMS algorithm (8). The purpose of such weighting is to attenuate frequency ranges in which the autocorrelation of the external input signal (51) is long, and thereby reduce the possibility of poorly adjusted coefficients and poor sound quality. Several possibilities exist for frequency weighting. Various combinations of fixed and adaptive filters have been suggested in the past.

20    In the present invention, steep highpass filters with high attenuation (20) are included in the inputs to the LMS algorithm. The purpose of these filters is to prevent low frequency contents from the reference signal (11) from entering the LMS algorithm. The cutoff frequency for the highpass filters (20) must be lower than the lowest frequency for which feedback cancellation should take place, and  
25    otherwise as high as possible.

With the highpass filters (20) in place, the LMS algorithm (8) would not experience an increased level of the error signal (10) when the coefficients (9) are poorly adjusted in the low frequency range. Filter (7) with poorly adjusted  
30    coefficients, combined with components (3) and (6), may lead to a system with a high loop gain, and instabilities may result.

In order to avoid this problem, a parallel feedback cancellation filter (21) is added. This filter is intended to provide low frequency information to the LMS

algorithm. The two filters (7) and (21) use identical coefficients (9). While filter (7) is designed to simulate the path consisting of components (4), (5), (1), (2) and (50), filter (21) is designed to simulate the artificial path (25) with an impulse response of constant '0'. The adder (33) computes an error signal as the difference between the desired '0' output and the actual output (34) from the filter (21). The error output (10) from the high frequency range and the error output (27) from the low frequency range are combined into a single error signal (28) which is fed to the error input of the LMS algorithm (8). In order to generate a low frequency signal as input to the filter (21) and to the reference input to the LMS algorithm, a noise generator (22) is included. The noise generator output (29) is lowpass filtered by a fixed filter (23). The cutoff frequency for the lowpass filter (23) is selected approximately equal to the cutoff frequency of the highpass filters (20), to obtain a reasonably flat input spectrum to the LMS algorithm. The low frequency signal (32) and the high frequency signal (31) are combined by the adder (24) to form the complete reference signal (30) for the LMS algorithm. Clearly, the components (25) and (33) may be removed immediately, and the signal (34) can be connected to the signal (27).

The noise generator (22) may be implemented by randomly swapping the numerical sign of each sample of the signal (35). In other words, for each sample instant it is randomly decided whether the sample value should be multiplied by 1 or by (-1). The advantage of using this type of noise generator is that noise samples at (35) and at (29) always have the same amplitude. The power spectrum of the reference signal (30) is therefore reasonably balanced at all times. In the literature, the noise generated as described above is sometimes referred to as 'Schroeder' noise.

#### **Feedback oscillation detector**

Feedback oscillations may be produced by a system which contains an amplifier and a feedback loop, under some circumstances. A hearing aid with acoustic amplification, combined with an acoustic path from the hearing aid telephone through a ventilation channel ("vent") and possibly other leaks, form a loop which may have a gain higher than 0 dB, at least for some frequencies. With more

than 0 dB loop gain, the system may become unstable and produce feedback oscillations.

5 The present invention is designed to detect a feedback oscillation in the input signal (55), and set a flag (15) which indicates 'oscillation' or 'no oscillation'.

10 Some assumptions about the feedback oscillations in hearing aids are included in the design of the detector. The signal produced as a feedback oscillation typically consists of a single frequency, namely the frequency at which the loop gain is highest, taking into account both the linear and non-linear components of the hearing aid. The level of the feedback oscillation is relatively stable, after a certain settling time. The feedback oscillation often dominates the signal in the feedback loop, since its level is often determined by the hearing aid compressors.

15 The feedback detection process is complicated by the presence of other signals in the feedback loop. Many environmental signals, including music, may contain segments of periodic nature which may resemble a feedback oscillation. However, in the frequency range where oscillations may occur, relatively few environmental signals consist of a single frequency only, at least when  
20 considered over a period of a few hundred milliseconds or more.

The feedback oscillation detector in the present invention is based on measuring the overall 'bandwidth' of the signal in the feedback loop consisting of components (1), (2), (3), (4), (5) and (50). In the preferred embodiment, the signal  
25 (55) is used as input to the detector, but with slight modifications the detector may obtain its input anywhere in the loop. When the bandwidth of the signal (55) has been small for a certain minimum period of time, the detector will flag a 'feedback oscillation' condition.

30 FIG. 3 describes the detector (49). The input signal (55) is highpass filtered by an 8-tap FIR filter (36). The filter helps prevent false feedback oscillation detection for low frequency input signals since it suppresses the fundamental frequencies for a wide range of signals. The 3 dB roll-off frequency for the filter should be higher than the lowest expected feedback oscillation frequency. The 8-tap FIR

filter is just one example of a usable filter, but many other types may be used. The highpass filtered signal (37) is fed to a modeling device (38), which attempts to model the spectrum of the signal (37), using a second-order auto-regressive model as shown in E4:

$$y(n) = x(n) \cdot K - a_1 y(n-1) - a_2 y(n-2) \quad (E4)$$

where  $x(n)$  represents the excitation signal, which drives the model input, while  $y(n)$  is the output from the model.

- 10 The signal model E4 represents a second-order IIR filter with a single complex-conjugated pole-pair. Based on the model coefficients  $a_1$  and  $a_2$  (39), the filters center frequency and bandwidth may be computed. This computation is performed by the unit (40), which produces a bandwidth (41) and a center frequency (48). These two values are compared by (47) to preset thresholds (43) and (46). The comparator sets flag (44) TRUE if the bandwidth (41) is lower than the preset threshold (43) AND the center frequency (48) is higher than the acceptable minimum feedback oscillation frequency (46). Otherwise the flag (44) is set FALSE.
- 20 All components (38), (40), (47) and (45) are working on a frame based schedule. A frame length of 40 ms may be used, but other values of the length would also work. For each frame, a new value of the flag (44) is computed. Since many environmental input signals contain short segments of narrow bandwidth, the flag (44) may occasionally be set TRUE while no feedback oscillations are present. To avoid this, the flag (44) is fed to a stability estimator (45). In here, the flag (44) is placed in a delay line which, at any point in time, holds the values of the flag from the last  $N_{ss}$  frames.  $N_{ss}$  may be selected as 10, but other values would also work. The stability estimator (45) sets the detector flag (15) TRUE when and only when at least  $N_{min}$  out of the  $N_{ss}$  past values of the flag (44) were TRUE. For example,  $N_{min}$  may be set to 4.
- 30

The coefficients  $a_1$  and  $a_2$  in E4 are computed from the autocorrelation coefficients  $R(0)$ ,  $R(1)$  and  $R(2)$ , by solving the equations:

$$R(0) \cdot a_1 + R(1) \cdot a_2 = -R(1)$$

(E5a)

$$R(1) \cdot a_1 + R(0) \cdot a_2 = -R(2)$$

(E5b)

The autocorrelation coefficients may be computed using the following equations:

$$R(0) = \frac{1}{N_f} \cdot \sum x(n)^2, \quad n = 1..N_f$$

(E6a)

$$R(1) = \frac{1}{N_f} \cdot \sum x(n) \cdot x(n+1), \quad n = 1..N_f - 1$$

(E6b)

$$R(2) = \frac{1}{N_f} \cdot \sum x(n) \cdot x(n+2), \quad n = 1..N_f - 2$$

(E6c)

where  $N_f$  corresponds to the frame length, and  $x(i)$  is the  $i$ 'th sample of signal (37) from the current frame.

The 3-dB bandwidth of the filter represented by the auto-regressive model E4 may be computed as

$$Bandwidth = 2 \cdot (1 - \sqrt{a_2})$$

(E7)

15

and the center frequency may be computed as

$$f_{Center} = \cos^{-1}\left(\frac{-a_1}{2\sqrt{a_2}}\right)$$

(E8)

In both equations (E7) and (E8) the result is given in radians. Simple calculations, in which the system sample rate is included, may be used to convert the values of Bandwidth and the  $f_{Center}$  into Hz.

20

In the previous description the hearing aid and the methods have been described in a simplified manner. Necessary elements like a power source, e.g. a battery, and related wiring, the signal processing capabilities of the hearing aid amplifier and the interconnecting wiring of the components, as well as the housing, which  
5 is always present have been omitted from the general definition of the hearing aid according to the invention. It should be appreciated that these elements of course are present in a hearing aid actually manufactured.

## CLAIMS

1. A method for canceling feedback in an acoustic system comprising a microphone, a signal path, a speaker, means for detecting presence of  
5 feedback between the speaker and the microphone and filter means for compensating at least partly a possible feedback signal, the method comprising:
  - providing a LMS algorithm for generating filter coefficients;
  - where the LMS algorithm operates with a predetermined essentially level  
10 independent adaptation speed when feedback is not present, this representing a first mode,
  - where the LMS algorithm operates a level dependent adaptation speed when feedback is present, this representing a second mode;
  - where the means for detecting the presence of feedback is used to control  
15 the adaptation mode selection of the LMS algorithm and
  - where the feedback detection means comprises bandwidth detection means for determining the presence of a feedback signal.
2. A method according to claim 1, where the update rate for the LMS algorithm  
20 is determined by the long-term average denominator in the LMS update algorithm in the second mode.
3. A method according to claim 1 or 2, comprising using a highpass filter to prevent low-frequency signals from entering the LMS algorithm; where an  
25 additional feedback cancellation filter and a noise generator is used for providing low-frequency input for the LMS algorithm.
4. A method according to claim 1, where the stability of the signal determined as  
a feedback signal is analyzed.
- 30 5. A method according to claim 4, where the feedback analyzing comprises holding flag values from a number of succeeding time frames and comparing of these.



6. A hearing aid comprising :
- a microphone;
  - a signal path;
  - a amplifier;
  - 5 - a speaker;
  - means for detecting feedback between the speaker and the microphone;
  - filter means for compensating at least partly a possible feedback signal;
  - memory means including a LMS algorithm;
  - means for shifting the adaptation mode of the LMS algorithm when
  - 10 feedback is detected, said means being controlled by the means for detecting feedback and
  - means for updating the LMS algorithm by the long term denominator in the LMS algorithm;
  - where the feedback detection means comprises bandwidth detection
  - 15 means for determining the presence of a feedback signal.
7. A hearing aid according to claim 6, comprising stability detecting means for the feedback signal.

1/3

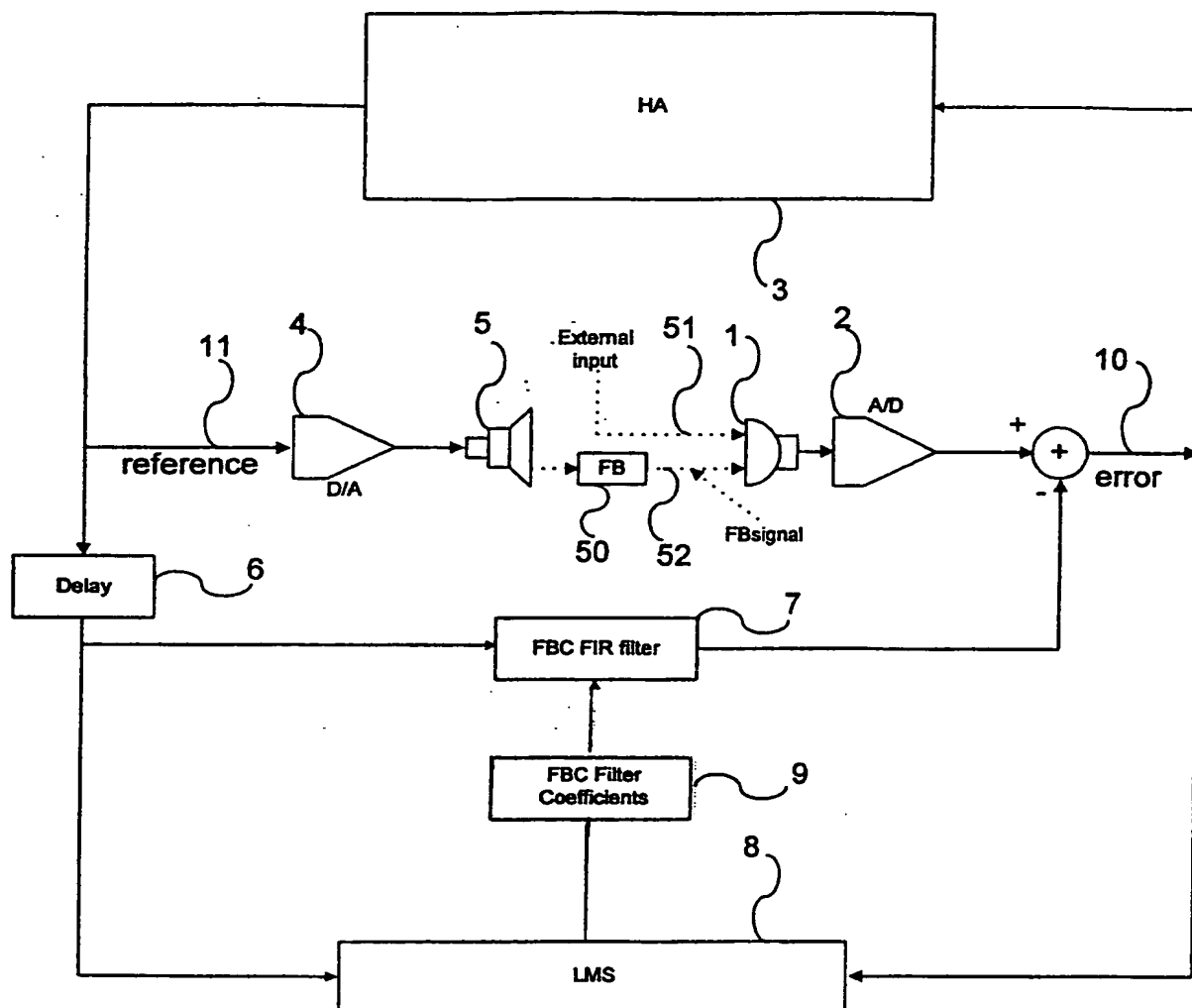


Fig. 1

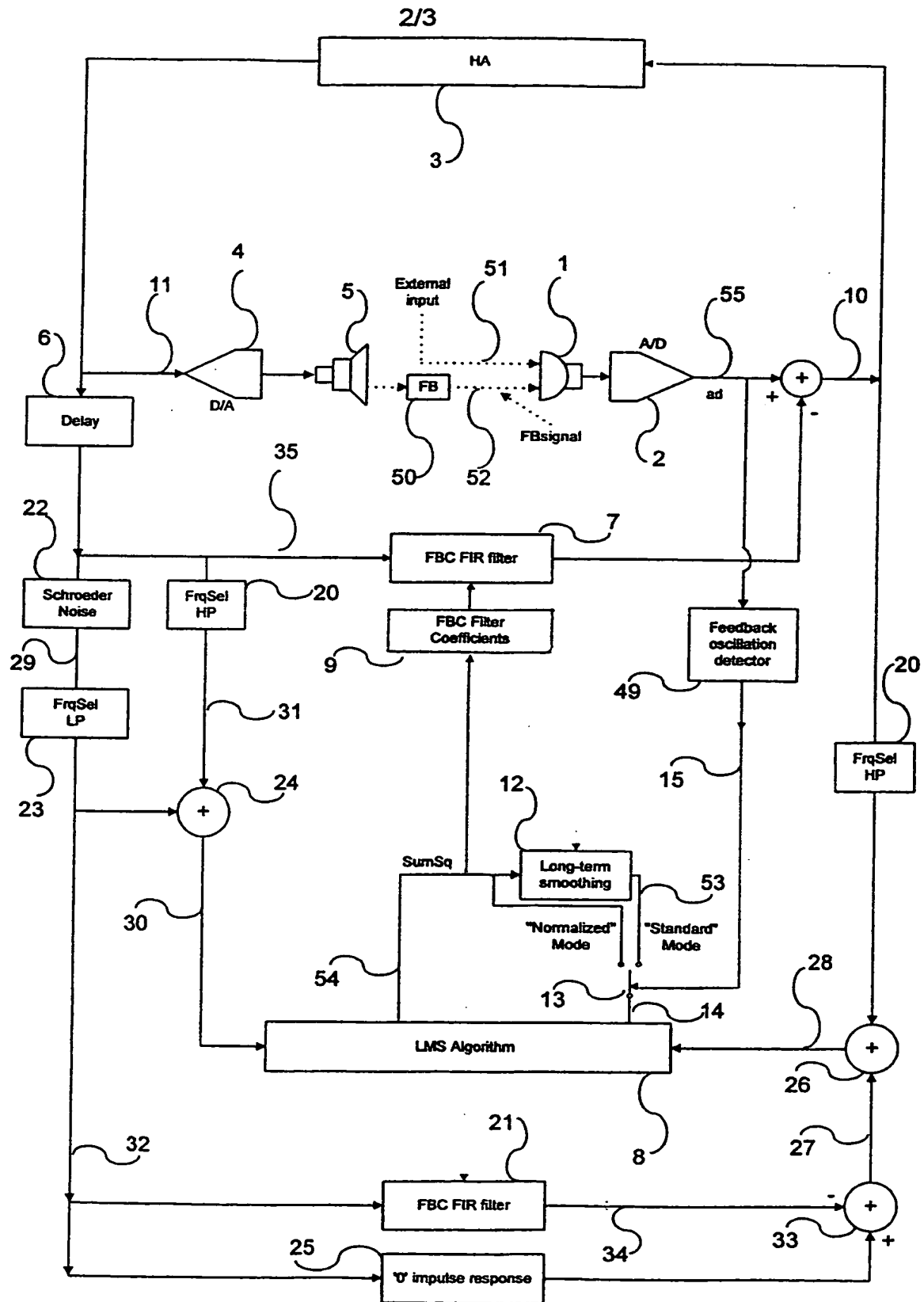


Fig. 2

3/3

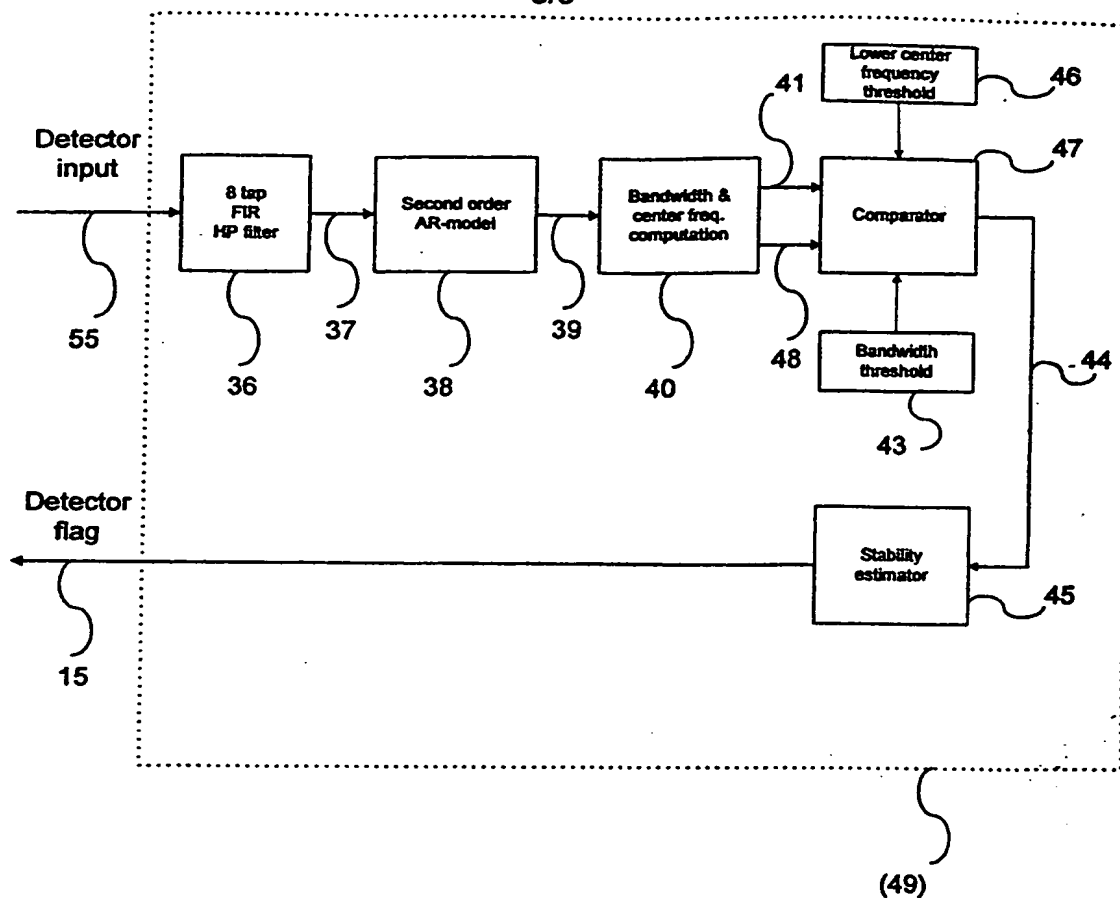


Fig. 3

(19) World Intellectual Property Organization  
International Bureau



(43) International Publication Date  
25 January 2001 (25.01.2001)

PCT

(10) International Publication Number  
**WO 01/06746 A3**

(51) International Patent Classification<sup>7</sup>: H04R 25/00, 3/02

(DK). EKELID, Michael [DK/DK]; Oticon A/S, Strandvejen 58, DK-2900 Hellerup (DK).

(21) International Application Number: PCT/DK00/00379

(74) Common Representative: OTICON A/S; Mikael, T., Christensen, Strandvejen 58, DK-2900 Hellerup (DK).

(22) International Filing Date: 7 July 2000 (07.07.2000)

(81) Designated States (*national*): AU, BR, CA, JP, US.

(25) Filing Language: English

(84) Designated States (*regional*): European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

(26) Publication Language: English

(30) Priority Data:  
PA 1999 01043 19 July 1999 (19.07.1999) DK

Published:  
— with international search report

(71) Applicant (*for all designated States except US*): OTICON A/S [DK/DK]; Strandvejen 58, DK-2900 Hellerup (DK).

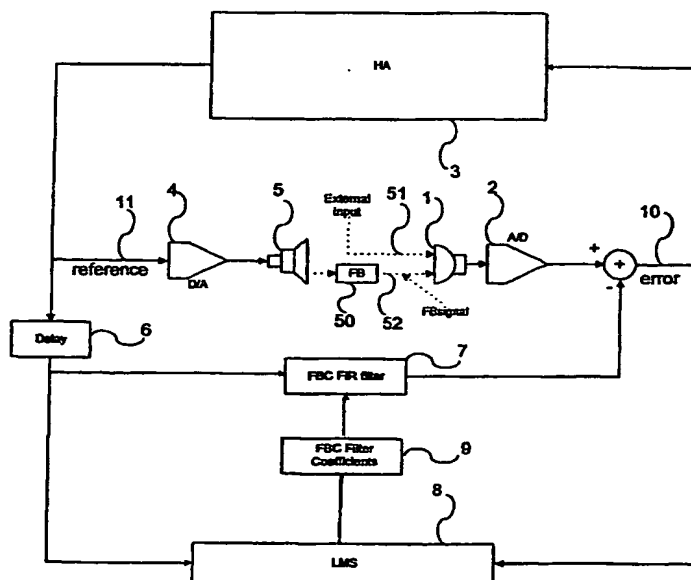
(88) Date of publication of the international search report:  
9 August 2001

(72) Inventors; and

(75) Inventors/Applicants (*for US only*): NIELSEN, Jakob [DK/DK]; Oticon A/S, Strandvejen 58, DK-2900 Hellerup

*For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.*

(54) Title: FEEDBACK CANCELLATION USING BANDWIDTH DETECTION



(57) Abstract: The invention relates to a method for cancelling feedback in an acoustic system comprising a microphone, a signal path, a speaker and means for detecting presence of feedback between the speaker and the microphone, the method comprising providing an LMS algorithm for processing the signal; where the LMS algorithm operates with a predetermined adaptation speed when feedback is not present; where the LMS algorithm operates an adaptation speed faster than the predetermined adaptation speed when feedback is present, and where the means for detecting the presence of feedback is used to control the adaptation speed selection of the LMS algorithm, where the feedback detection means comprises bandwidth detection means for determining the presence of a feedback signal.

WO 01/06746 A3

# INTERNATIONAL SEARCH REPORT

Int. Application No  
PCT/DK 00/00379

**A. CLASSIFICATION OF SUBJECT MATTER**  
IPC 7 H04R25/00 H04R3/02

According to International Patent Classification (IPC) or to both national classification and IPC

## B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 H04R

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

## C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 93 20668 A (GN DANAVOX AS ;HANSEN ROY SKOVGAARD (DK)) 14 October 1993 (1993-10-14) page 14, line 31 -page 15, line 25; figures	1-7
A	EP 0 581 261 A (MINNESOTA MINING & MFG) 2 February 1994 (1994-02-02) column 12, line 13 -column 13, line 4; figures	1-7

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

### \* Special categories of cited documents :

- \*A\* document defining the general state of the art which is not considered to be of particular relevance
- \*E\* earlier document but published on or after the international filing date
- \*L\* document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
- \*O\* document referring to an oral disclosure, use, exhibition or other means
- \*P\* document published prior to the international filing date but later than the priority date claimed

- \*T\* later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
- \*X\* document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
- \*Y\* document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
- \*G\* document member of the same patent family

Date of the actual completion of the international search

14 March 2001

Date of mailing of the international search report

20/03/2001

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2  
NL - 2280 HV Rijswijk  
Tel. (+31-70) 340-2040. Tx. 31 651 epo nl.  
Fax: (+31-70) 340-3016

Authorized officer

Gastaldi, G

# INTERNATIONAL SEARCH REPORT

Information on patent family members

Int. Application No

PCT/DK 00/00379

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9320668 A	14-10-1993	DK- 43292 A	01-10-1993
		AU 3948293 A	08-11-1993
		DE 69327951 D	06-04-2000
		DE 69327951 T	17-08-2000
		EP 0634084 A	18-01-1995
		JP 7505271 T	08-06-1995
		US 5680467 A	21-10-1997
EP 0581261 A	02-02-1994	AU 4183293 A	03-02-1994
		CA 2100015 A	30-01-1994
		DE 69326510 D	28-10-1999
		DE 69326510 T	18-05-2000
		DK 581261 T	10-04-2000
		JP 6189397 A	08-07-1994

REC'D 12 NOV 2001

WIPO PCT

## INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

5

Applicant's or agent's file reference P-1999-014-1	<b>FOR FURTHER ACTION</b> See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)	
International application No. PCT/DK00/00379	International filing date (day/month/year) 07/07/2000	Priority date (day/month/year) 19/07/1999
International Patent Classification (IPC) or national classification and IPC H04M9/00		
Applicant OTICON A/S et al.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.



2. This REPORT consists of a total of 5 sheets, including this cover sheet.

- ☐ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

- I ☒ Basis of the report
- II ☐ Priority
- III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
- IV ☐ Lack of unity of invention
- V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
- VI ☒ Certain documents cited
- VII ☒ Certain defects in the international application
- VIII ☒ Certain observations on the international application

Date of submission of the demand  19/02/2001	Date of completion of this report  07.11.2001
Name and mailing address of the international preliminary examining authority:   European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer  Tillgren, M  Telephone No. +49 89 2399 7497 



# INTERNATIONAL PRELIMINARY EXAMINATION REPORT

International application No. PCT/DK00/00379

## I. Basis of the report

1. With regard to the **elements** of the international application (*Replacement sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to this report since they do not contain amendments (Rules 70.16 and 70.17)*):

**Description, pages:**

1-13 as originally filed

**Claims, No.:**

1-7 as originally filed

**Drawings, sheets:**

1/3-3/3 as originally filed

2. With regard to the **language**, all the elements marked above were available or furnished to this Authority in the language in which the international application was filed, unless otherwise indicated under this item.

These elements were available or furnished to this Authority in the following language: , which is:

- ☐ the language of a translation furnished for the purposes of the international search (under Rule 23.1(b)).
- ☐ the language of publication of the international application (under Rule 48.3(b)).
- ☐ the language of a translation furnished for the purposes of international preliminary examination (under Rule 55.2 and/or 55.3).

3. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international preliminary examination was carried out on the basis of the sequence listing:

- ☐ contained in the international application in written form.
- ☐ filed together with the international application in computer readable form.
- ☐ furnished subsequently to this Authority in written form.
- ☐ furnished subsequently to this Authority in computer readable form.
- ☐ The statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.
- ☐ The statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished.

4. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:

**INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT**

International application No. PCT/DK00/00379

☐ the drawings, sheets:

5. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

*(Any replacement sheet containing such amendments must be referred to under item 1 and annexed to this report.)*

6. Additional observations, if necessary:

**V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement**

**1. Statement**

Novelty (N)	Yes:	Claims	1-7
	No:	Claims	
Inventive step (IS)	Yes:	Claims	1-7
	No:	Claims	
Industrial applicability (IA)	Yes:	Claims	1-7
	No:	Claims	

2. Citations and explanations  
**see separate sheet**

**VII. Certain defects in the international application**

The following defects in the form or contents of the international application have been noted:  
**see separate sheet**

**VIII. Certain observations on the international application**

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

**see separate sheet**

**Re Item V**

- 1) With regard to independent claim 1 the document EP-A-0 581 261 (hereinafter referred to as D1), which is regarded as closest prior art, discloses a method for cancelling feedback in an acoustic system (see abstract), comprising a microphone (column 1, lines 15-20), a signal path (see the drawing of the abstract), a speaker (column 1, lines 15-20), filter means for compensating at least partly a possible feedback signal (column 1, lines 39-47), wherein the method comprises, providing a LMS algorithm to generating filter coefficients (column 12, lines 13-26), where the LMS algorithm operates with a predetermined essentially level independent adaptation speed when feedback is not present, this representing a first mode (column 13, lines 22- 28), where the LMS algorithm operates at a level dependent adaptation speed when feedback is present, this representing a second mode (column 12, line 30- column 13, line 5).

The difference between the method of claim 1 and that of D1 is that the change between modes is initiated manually instead of by means of a means for detecting the presence of feedback. The objective problem to be solved is therefore how to make the manual process of detecting feedback and changing mode into an automatic one. To detect and change mode automatically is known from the document WO-A-93 20668 (hereinafter referred to as D2) (see for instance page 2, lines 18-33).

The document D2 does however not show the use of a bandwidth detection means for detecting a feedback signal, instead a detection form is used that will detect any change to the feedback-path. Since the detection of D2 might detect other changes than feedback signals this might lead to false detections and therefore annoying noise. Since there is nothing in the prior art that would lead the man skilled in the art to consider a method like the one in claim 1, it is considered to be novel an inventive.

Hence claim 1 does fulfill the requirements of Article 33(1)-(3) PCT since its subject matter is novel and does involve an inventive step.

- 2) Due to their dependencies claims 2-5 also fulfill the requirements of Article 33(1)-(3) PCT.

- 3) Regarding independent claim 6 its subject matter, with the exception that it is a hearing aid and not an acoustic system in general, the same as the subject matter of claim 2 (when read in conjunction with claim 1).

Hence claim 6 does fulfill the requirements of Article 33(1)-(3) PCT since its subject matter is novel and does involve an inventive step.

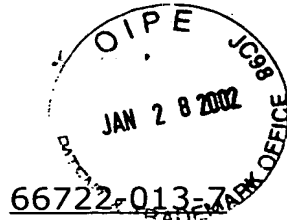
- 4) Due to its dependency claims 7 also fulfill the requirements of Article 33(1)-(3) PCT.

**Re Item VII**

- 1) Independent claims 1 and 6 are not in the two-part form in accordance with Rule 6.3(b) PCT, which in the present case would be appropriate, with known features (see D1 and D2) being placed in the preamble (Rule 6.3(b)(i) PCT) and with the remaining features being included in the characterising part (Rule 6.3(b)(ii) PCT). Said claims should therefore have been redrafted accordingly.
- 2) Reference signs in parentheses should have been inserted in the claims to increase their intelligibility (Rule 6.2(b) PCT). This also applies to method claims in as far as they refer to apparatus or flow chart features.
- 3) To meet the requirements of Rule 5.1(a)(ii) PCT, the documents D1 and D2 should have been identified in the description and the relevant background art disclosed therein should have been discussed.

**Re Item VIII**

- 1) Claims 1-5 are not supported by the description as required by Article 6 PCT, as their scope is broader than justified by the description and drawings. The reasons therefor are the following: The description and the drawings are all aimed towards a hearing aid, whereas claims 1-5 are all directed towards an acoustic system in general. Since there is no reference made in the description to a general acoustic system said claims are unduly broad and should therefore have been limited to consider only a hearing aid.



10/08/337

JC10 Res. PCI/PTO 28 JAN 2002

2643 #1  
sm  
6/23/02

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of: ) PATENT  
)  
Jakob NIELSEN et al. ) Group Art Unit: Unknown  
)  
Serial No.: (PCT/DK00/00379) ) Examiner: Unknown  
)  
Filed: (7 July 2000) )  
)  
FEEDBACK CANCELLATION USING )  
BANDWIDTH DETECTION )

RECEIVED

JUN 11 2002

Technology Center 2600

\* \* \* \* \*

SUBMISSION OF INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT

Washington, D.C.  
January 28, 2002

Honorable Director for Patents  
Washington, D.C. 20231

Sir:

The applicants submit a copy of the International Preliminary  
Examination Report, issued 7 November 2001.

Respectfully submitted,

**DYKEMA GOSSETT PLLC**

Richard H. Tushin  
Registration No. 27,297  
Franklin Square, Third Floor West  
1300 I Street, NW  
Washington, DC 20005-3306  
(202) 906-8680



RECEIVED  
JUN 10 2002  
TC 2800 MAIL ROOM

# PATENT COOPERATION TREATY

## PCT

### INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference <b>P-1999-014-1</b>		<b>FOR FURTHER ACTION</b> See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)	
International application No. <b>PCT/DK00/00379</b>	International filing date (day/month/year) <b>07/07/2000</b>	Priority date (day/month/year) <b>19/07/1999</b>	
International Patent Classification (IPC) or national classification and IPC <b>H04M9/00</b>			
Applicant <b>OTICON A/S et al.</b>			
<p>1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.</p> <p>2. This REPORT consists of a total of 5 sheets, including this cover sheet.</p> <p><input type="checkbox"/> This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).</p> <p>These annexes consist of a total of sheets.</p>			
<p>3. This report contains indications relating to the following items:</p> <ul style="list-style-type: none"> <li>I <input checked="" type="checkbox"/> Basis of the report</li> <li>II <input type="checkbox"/> Priority</li> <li>III <input type="checkbox"/> Non-establishment of opinion with regard to novelty, inventive step and industrial applicability</li> <li>IV <input type="checkbox"/> Lack of unity of invention</li> <li>V <input checked="" type="checkbox"/> Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement</li> <li>VI <input checked="" type="checkbox"/> Certain documents cited</li> <li>VII <input checked="" type="checkbox"/> Certain defects in the international application</li> <li>VIII <input type="checkbox"/> Certain observations on the international application</li> </ul>			
Date of submission of the demand <b>19/02/2001</b>		Date of completion of this report <b>07.11.2001</b>	
Name and mailing address of the international preliminary examining authority:  <b>European Patent Office</b> <b>D-80298 Munich</b> <b>Tel. +49 89 2399 - 0 Tx: 523656 eprmu d</b> <b>Fax: +49 89 2399 - 4465</b>		Authorized officer  <b>Tillgren, M</b>  Telephone No. <b>+49 89 2399 7497</b> 	

**INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT**

International application No. PCT/DK00/00379

**I. Basis of the report**

1. With regard to the **elements** of the international application (*Replacement sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to this report since they do not contain amendments (Rules 70.16 and 70.17)*):

**Description, pages:**

1-13 as originally filed

**Claims, No.:**

1-7 as originally filed

**Drawings, sheets:**

1/3-3/3 as originally filed

2. With regard to the **language**, all the elements marked above were available or furnished to this Authority in the language in which the international application was filed, unless otherwise indicated under this item.

These elements were available or furnished to this Authority in the following language: , which is:

- ☐ the language of a translation furnished for the purposes of the international search (under Rule 23.1(b)).
- ☐ the language of publication of the international application (under Rule 48.3(b)).
- ☐ the language of a translation furnished for the purposes of international preliminary examination (under Rule 55.2 and/or 55.3).

3. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international preliminary examination was carried out on the basis of the sequence listing:

- ☐ contained in the international application in written form.
- ☐ filed together with the international application in computer readable form.
- ☐ furnished subsequently to this Authority in written form.
- ☐ furnished subsequently to this Authority in computer readable form.
- ☐ The statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.
- ☐ The statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished.

4. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:

**INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT**

International application No. PCT/DK00/00379

☐ the drawings, sheets:

5. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

*(Any replacement sheet containing such amendments must be referred to under item 1 and annexed to this report.)*

6. Additional observations, if necessary:

**V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement**

**1. Statement**

Novelty (N)	Yes:	Claims	1-7
	No:	Claims	
Inventive step (IS)	Yes:	Claims	1-7
	No:	Claims	
Industrial applicability (IA)	Yes:	Claims	1-7
	No:	Claims	

2. Citations and explanations  
**see separate sheet**

**VI. Certain documents cited**

1. Certain published documents (Rule 70.10)

and / or

2. Non-written disclosures (Rule 70.9)

**see separate sheet**

**VII. Certain defects in the international application**

The following defects in the form or contents of the international application have been noted:  
**see separate sheet**



**Re Item V**

- 1) With regard to independent claim 1 the document EP-A-0 581 261 (hereinafter referred to as D1), which is regarded as closest prior art, discloses a method for cancelling feedback in an acoustic system (see abstract), comprising a microphone (column 1, lines 15-20), a signal path (see the drawing of the abstract), a speaker (column 1, lines 15-20), filter means for compensating at least partly a possible feedback signal (column 1, lines 39-47), wherein the method comprises, providing a LMS algorithm to generating filter coefficients (column 12, lines 13-26), where the LMS algorithm operates with a predetermined essentially level independent adaptation speed when feedback is not present, this representing a first mode (column 13, lines 22- 28), where the LMS algorithm operates at a level dependent adaptation speed when feedback is present, this representing a second mode (column 12, line 30- column 13, line 5).

The difference between the method of claim 1 and that of D1 is that the change between modes is initiated manually instead of by means of a means for detecting the presence of feedback. The objective problem to be solved is therefore how to make the manual process of detecting feedback and changing mode into an automatic one. To detect and change mode automatically is known from the document WO-A-93 20668 (hereinafter referred to as D2) (see for instance page 2, lines 18-33).

The document D2 does however not show the use of a bandwidth detection means for detecting a feedback signal, instead a detection form is used that will detect any change to the feedback-path. Since the detection of D2 might detect other changes than feedback signals this might lead to false detections and therefore annoying noise. Since there is nothing in the prior art that would lead the man skilled in the art to consider a method like the one in claim 1, it is considered to be novel an inventive.

Hence claim 1 does fulfill the requirements of Article 33(1)-(3) PCT since its subject matter is novel and does involve an inventive step.

- 2) Due to their dependencies claims 2-5 also fulfill the requirements of Article 33(1)-(3) PCT.

**INTERNATIONAL PRELIMINARY  
EXAMINATION REPORT - SEPARATE SHEET**

---

International application No. PCT/DK00/00379

- 3) Regarding independent claim 6 its subject matter, with the exception that it is a hearing aid and not an acoustic system in general, the same as the subject matter of claim 2 (when read in conjunction with claim 1).

Hence claim 6 does fulfill the requirements of Article 33(1)-(3) PCT since its subject matter is novel and does involve an inventive step.

- 4) Due to its dependency claims 7 also fulfill the requirements of Article 33(1)-(3) PCT.

**Re Item VII**

- 1) Independent claims 1 and 6 are not in the two-part form in accordance with Rule 6.3(b) PCT, which in the present case would be appropriate, with known features (see D1 and D2) being placed in the preamble (Rule 6.3(b)(i) PCT) and with the remaining features being included in the characterising part (Rule 6.3(b)(ii) PCT). Said claims should therefore have been redrafted accordingly.
- 2) Reference signs in parentheses should have been inserted in the claims to increase their intelligibility (Rule 6.2(b) PCT). This also applies to method claims in as far as they refer to apparatus or flow chart features.
- 3) To meet the requirements of Rule 5.1(a)(ii) PCT, the documents D1 and D2 should have been identified in the description and the relevant background art disclosed therein should have been discussed.

**Re Item VIII**

- 1) Claims 1-5 are not supported by the description as required by Article 6 PCT, as their scope is broader than justified by the description and drawings. The reasons therefor are the following: The description and the drawings are all aimed towards a hearing aid, whereas claims 1-5 are all directed towards an acoustic system in general. Since there is no reference made in the description to a general acoustic system said claims are unduly broad and should therefore have been limited to consider only a hearing aid.

# PATENT COOPERATION TREATY

**PCT**

## NOTIFICATION OF ELECTION

(PCT Rule 61.2)

From the INTERNATIONAL BUREAU

To:

Commissioner  
US Department of Commerce  
United States Patent and Trademark  
Office, PCT  
2011 South Clark Place Room  
CP2/5C24  
Arlington, VA 22202  
ETATS-UNIS D'AMERIQUE

in its capacity as elected Office

<b>Date of mailing (day/month/year)</b> 02 April 2001 (02.04.01)	<b>Applicant's or agent's file reference</b> P-1999-014-1
<b>International application No.</b> PCT/DK00/00379	<b>Priority date (day/month/year)</b> 19 July 1999 (19.07.99)
<b>International filing date (day/month/year)</b> 07 July 2000 (07.07.00)	
<b>Applicant</b> NIELSEN, Jakob et al	

1. The designated Office is hereby notified of its election made:

☒ in the demand filed with the International Preliminary Examining Authority on:

19 February 2001 (19.02.01)

☐ in a notice effecting later election filed with the International Bureau on:

2. The election ☒ was

☐ was not

made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

<b>The International Bureau of WIPO</b> 34, chemin des Colombettes 1211 Geneva 20, Switzerland	<b>Authorized officer</b>  Claudio Borton
Facsimile No.: (41-22) 740.14.35	Telephone No.: (41-22) 338.83.38